Caller ID Delivery in AOS Voice Devices

When a call is destined for a Voice User or Trunk on the AOS voice device, CID is delivered in different ways.

- When CID is delivered to an analog Voice User it is sent via FSK.
- When CID is delivered to a SIP Voice User it is sent in the FROM: header in the SIP message.
- When CID is delivered to a PRI it is sent in the ISDN Setup message or the subsequent Facility message.
- When CID is delivered to a CAS FGD E&M Trunk it is sent via DTMF tones.

It is important to note that CID information must be properly delivered to the AOS voice device so it can be passed on. If there is no useful CID information to extract from a SIP INVITE received by the ADTRAN, then no useful CID information can be passed on to the end user behind the ADTRAN. Likewise, if the AOS voice device does not receive CID information from a PBX, then it will not generate CID information in the SIP messages sent out towards the network. In this latter case, CID override commands will need to be set in the ADTRAN. Please reference Knowledgebase Article #2009 for information on manipulating CID. This document covers the delivery of CID and is current as of Feb. 2008, AOS 16.04.00.SB.

CID to an analog Voice User

The AOS voice device will send CID (ANI) information via Frequency Shift Keying (FSK) out the analog FXS port to a phone or PBX, 500ms after the first ring. FSK is a type of tone modulation that is different than DTMF (Dual-Tone Multi-Frequency). The CID format supported is Multiple Data Message Format (MDMF), which contains name, number, time, and date. Caller ID to an analog Voice User can be debugged in the AOS voice device by the following command:

debug voice toneservices

A typical debug looks like this:

```
16:42:35 TONESERVICES.EVENTS fxs 0/2 - empty - Caller-ID Generation: Request resource
16:42:35 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Caller-ID Generation: DSP channel allocated for the resource
16:42:35 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Caller-ID Generation: constructed
16:42:37 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Caller-ID Generation: starting Call-Waiting Caller-ID alert and maybe sending Caller-ID:
16:42:37 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Caller-ID Generation:   chars = "....02191642..5559075..First Last"
16:42:37 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Caller-ID Generation:   bytes = "80 1F 01 08 30 32 31 39 31 36 34 32 02 07 35 35 35 39 30 37 35 07 0A 46 69 72 73 74 20 4C 61 73 74 75"
16:42:37 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Caller-ID Generation: TDM map
16:42:39 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Caller-ID Generation: received Caller-ID Done event
16:42:39 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Caller-ID Generation: stopping
16:42:39 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Caller-ID Generation: TDM unmap
16:42:39 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Caller-ID Generation: release
```

The 6th line in this debug is the easiest to read. The date, time, number, and name are from line 6 are shown below:

"..02191642..5559075..First Last"

0219 – Date　　1642 – Time　　5559075 - calling party number　　First Last – calling party name

In addition to '**voice toneservices**', '**debug interface fxs**' can also be used to see off-hook, on-hook, and ring-cycle events.

CID to a SIP Voice User

The AOS voice device will send CID information to a SIP phone/device in the FROM: field in a SIP message. The FROM: field contains name and number. The time can be seen in the timestamp off to the left-hand side. The Time and Date should be set in the ADTRAN in order to deliver an accurate time to the end user. The Time and Date can be set two ways:

set the AOS voice device to use an SNTP server

sntp server 10.1.1.1 <ip address>

set the clock manually

clock set <HH:MM:SS>

Caller ID to a Sip phone/device can be debugged in the AOS voice device by the following command:

debug sip stack message

A typical debug looks like this:

```
11:41:09 SIP.STACK MSG    Tx: UDP src=10.10.10.21:5060 dst=10.1.4.5:5060
11:41:09 SIP.STACK MSG      INVITE sip555:9075@10.1.1.5:5060 SIP/2.0
11:41:09 SIP.STACK MSG      From: "First Last"<sip:5559077@10.1.1.5:5060;transport=UDP>;tag=2fab390-a13d579-13c4-
3a111f-27ef8d7f-3a111f
11:41:09 SIP.STACK MSG      To: <sip:5559075@10.1.1.5:5060;transport=UDP>
11:41:09 SIP.STACK MSG      Call-ID: 2ff3b90-a13d579-13c4-3a111f-2515461e-3a111f@10.1.4.5
11:41:09 SIP.STACK MSG      CSeq: 1 INVITE
11:41:09 SIP.STACK MSG      Via: SIP/2.0/UDP 10.19.213.121:5060;branch=z9hG4bK-3a111f-e2d2e194-6f1c579c
11:41:09 SIP.STACK MSG      Max-Forwards: 70
11:41:09 SIP.STACK MSG      Supported: 100rel,replaces
11:41:09 SIP.STACK MSG      Allow: ACK, BYE, CANCEL, INFO, INVITE, NOTIFY, OPTIONS, PRACK, REFER, REGISTER
11:41:09 SIP.STACK MSG      User-Agent: ADTRAN_Total_Access_916e/16.03.00.E
11:41:09 SIP.STACK MSG      Contact: <sip:5559077@10.10.10.21:5060;transport=UDP>
11:41:09 SIP.STACK MSG      Content-Type: application/SDP
11:41:09 SIP.STACK MSG      Content-Length: 266
11:41:09 SIP.STACK MSG
11:41:09 SIP.STACK MSG      v=0
11:41:09 SIP.STACK MSG      o=- 1203529269 1203529269 IN IP4 10.10.10.21
11:41:09 SIP.STACK MSG      s=-
11:41:09 SIP.STACK MSG      c=IN IP4 10.10.10.21
11:41:09 SIP.STACK MSG      t=0 0
```

```
11:41:09 SIP.STACK MSG       m=audio 10048 RTP/AVP 0 18 101
11:41:09 SIP.STACK MSG       a=rtpmap:0 PCMU/8000
11:41:09 SIP.STACK MSG       a=rtpmap:18 G729/8000
11:41:09 SIP.STACK MSG       a=fmtp:18 annexb=no
11:41:09 SIP.STACK MSG       a=silenceSupp:off - - - -
11:41:09 SIP.STACK MSG       a=rtpmap:101 telephone-event/8000
11:41:09 SIP.STACK MSG       a=fmtp:101 0-15
```

The 3rd line in this debug contains the FROM: field. In this field, the display name and calling party number can be seen.

CID to a PRI Trunk

The AOS voice device will send CID information to a PBX connected via PRI in the Setup message or a subsequent Facility message. Commands in the PRI interface in the AOS voice device allow the user to specify whether or not CID name appears in the Setup message. Caller ID to a PRI can be debugged in the AOS voice device by the following command:

debug isdn l2-formatted

A typical debug for CID Name delivered in the Setup message looks like this:

```
13:58:26 ISDN.L2_FMT PRI  1  ====================================================
13:58:26 ISDN.L2_FMT PRI  1  Sent = Sapi:00  C/R:C Tei:00
13:58:26 ISDN.L2_FMT PRI  1       Ctl:INFO     Ns:5   Nr:5
13:58:26 ISDN.L2_FMT PRI  1       Prot:08  CRL:2  CRV:000F
13:58:26 ISDN.L2_FMT PRI  1       M - 05 SETUP
13:58:26 ISDN.L2_FMT PRI  1        IE - 04 BEARER CAPABILITY   Len=3
13:58:26 ISDN.L2_FMT PRI  1            80 Xfer Cap.:SPEECH
13:58:26 ISDN.L2_FMT PRI  1            90 Xfer Rate:64k
13:58:26 ISDN.L2_FMT PRI  1            A2 Layer 1:u-Law
13:58:26 ISDN.L2_FMT PRI  1        IE - 18 CHANNEL ID         Len=3
13:58:26 ISDN.L2_FMT PRI  1            A1 Primary Rate
13:58:26 ISDN.L2_FMT PRI  1               Intfc ID:IMPLICIT
13:58:26 ISDN.L2_FMT PRI  1               Pref/Excl:PREFERRED
13:58:26 ISDN.L2_FMT PRI  1               D-Chan Indicated:NO
13:58:26 ISDN.L2_FMT PRI  1               Chan. Sel:FOLLOWS
13:58:26 ISDN.L2_FMT PRI  1            83 Numb/Map:NUMBER
13:58:26 ISDN.L2_FMT PRI  1            89 Channel:9
13:58:26 ISDN.L2_FMT PRI  1        IE - 1C FACILITY          Len=24
13:58:26 ISDN.L2_FMT PRI  1         Calling Name: First Last
13:58:26 ISDN.L2_FMT PRI  1        IE - 6C CALLING PARTY #     Len=6
13:58:26 ISDN.L2_FMT PRI  1            00 Numb. Type:UNKNOWN
13:58:26 ISDN.L2_FMT PRI  1               Numb. Plan:UNKNOWN
13:58:26 ISDN.L2_FMT PRI  1            80 Presentation:ALLOWED
13:58:26 ISDN.L2_FMT PRI  1               Ph.# 5559077
13:58:26 ISDN.L2_FMT PRI  1        IE - 70 CALLED PARTY #      Len=5
13:58:26 ISDN.L2_FMT PRI  1            80 Numb. Type:UNKNOWN
13:58:26 ISDN.L2_FMT PRI  1               Numb. Plan:UNKNOWN
13:58:26 ISDN.L2_FMT PRI  1               Ph.# 5559078
13:58:26 ISDN.L2_FMT PRI  1  ====================================================
```

The above Setup message contains both CID name and number. The name is found in the Facility Information Element (IE) above the Calling Party number information. An option can be set in the PRI interface to display the CID name in a Display IE (usually for DMS 100). The command below shows how the CID name is presents:

isdn name-delivery <display | proceeding | setup>

A typical debug for CID Name delivered after the Call Proceeding message looks like this:

```
TA924e(config-pri 1)#
14:30:07 ISDN.L2_FMT PRI  1  ====================================================
14:30:07 ISDN.L2_FMT PRI  1  Sent = Sapi:00  C/R:C Tei:00
14:30:07 ISDN.L2_FMT PRI  1        Ctl:INFO     Ns:16   Nr:24
14:30:07 ISDN.L2_FMT PRI  1        Prot:08  CRL:2  CRV:0009
14:30:07 ISDN.L2_FMT PRI  1        M - 05 SETUP
14:30:07 ISDN.L2_FMT PRI  1         IE - 04 BEARER CAPABILITY   Len=3
14:30:07 ISDN.L2_FMT PRI  1             80 Xfer Cap.:SPEECH
14:30:07 ISDN.L2_FMT PRI  1             90 Xfer Rate:64k
14:30:07 ISDN.L2_FMT PRI  1             A2 Layer 1:u-Law
14:30:07 ISDN.L2_FMT PRI  1         IE - 18 CHANNEL ID        Len=3
14:30:07 ISDN.L2_FMT PRI  1             A1 Primary Rate
14:30:07 ISDN.L2_FMT PRI  1                Intfc ID:IMPLICIT
14:30:07 ISDN.L2_FMT PRI  1                Pref/Excl:PREFERRED
14:30:07 ISDN.L2_FMT PRI  1                D-Chan Indicated:NO
14:30:07 ISDN.L2_FMT PRI  1                Chan. Sel:FOLLOWS
14:30:07 ISDN.L2_FMT PRI  1             83 Numb/Map:NUMBER
14:30:07 ISDN.L2_FMT PRI  1             8f Channel:15
14:30:07 ISDN.L2_FMT PRI  1         IE - 6C CALLING PARTY #     Len=6
14:30:07 ISDN.L2_FMT PRI  1             00 Numb. Type:UNKNOWN
14:30:07 ISDN.L2_FMT PRI  1                Numb. Plan:UNKNOWN
14:30:07 ISDN.L2_FMT PRI  1             80 Presentation:ALLOWED
14:30:07 ISDN.L2_FMT PRI  1                Ph.# 5559077
14:30:07 ISDN.L2_FMT PRI  1         IE - 70 CALLED PARTY #      Len=5
14:30:07 ISDN.L2_FMT PRI  1             80 Numb. Type:UNKNOWN
14:30:07 ISDN.L2_FMT PRI  1                Numb. Plan:UNKNOWN
14:30:07 ISDN.L2_FMT PRI  1                Ph.# 5559078
14:30:07 ISDN.L2_FMT PRI  1  ====================================================
14:30:07 ISDN.L2_FMT PRI  1  Recd = Sapi:00  C/R:R Tei:00
14:30:07 ISDN.L2_FMT PRI  1        Ctl:INFO     Ns:24   Nr:17
14:30:07 ISDN.L2_FMT PRI  1        Prot:08  CRL:2  CRV:8009
14:30:07 ISDN.L2_FMT PRI  1        M - 02 CALL_PROC
14:30:07 ISDN.L2_FMT PRI  1         IE - 18 CHANNEL ID        Len=3
14:30:07 ISDN.L2_FMT PRI  1             A9 Primary Rate
14:30:07 ISDN.L2_FMT PRI  1                Intfc ID:IMPLICIT
14:30:07 ISDN.L2_FMT PRI  1                Pref/Excl:EXCLUSIVE
14:30:07 ISDN.L2_FMT PRI  1                D-Chan Indicated:NO
14:30:07 ISDN.L2_FMT PRI  1                Chan. Sel:FOLLOWS
14:30:07 ISDN.L2_FMT PRI  1             83 Numb/Map:NUMBER
14:30:07 ISDN.L2_FMT PRI  1             8f Channel:15
14:30:07 ISDN.L2_FMT PRI  1  ====================================================
14:30:07 ISDN.L2_FMT PRI  1  Recd = Sapi:00  C/R:R Tei:00
14:30:07 ISDN.L2_FMT PRI  1        Ctl:INFO     Ns:25   Nr:17
14:30:07 ISDN.L2_FMT PRI  1        Prot:08  CRL:2  CRV:8009
14:30:07 ISDN.L2_FMT PRI  1        M - 01 ALERTING
14:30:07 ISDN.L2_FMT PRI  1  ====================================================
14:30:07 ISDN.L2_FMT PRI  1  Sent = Sapi:00  C/R:C Tei:00
14:30:07 ISDN.L2_FMT PRI  1        Ctl:INFO     Ns:17   Nr:26
14:30:07 ISDN.L2_FMT PRI  1        Prot:08  CRL:2  CRV:0009
14:30:07 ISDN.L2_FMT PRI  1        M - 62 FACILITY
14:30:07 ISDN.L2_FMT PRI  1         IE - 1C FACILITY          Len=28
14:30:07 ISDN.L2_FMT PRI  1          Calling Name: First Last
```

The above Setup message contains only CID number information. The CID name appears in a Facility message after the Setup message.

CID to a CAS FGD E&M Trunk

The AOS voice device will send ANI to a Feature Group D (FGD) trunk via DTMF tones, after receiving a wink from the PBX. CID cannot be sent over standard E&M trunks because there is no true ring cadence like there is in Loop Start. FGD is very similar to E&M Wink, however when the PBX winks back toward the ADTRAN, the ADTRAN sends more information than just Called Party digits (DNIS). *Note: The AOS voice device does not support MF, only DTMF. If the PBX only supports MF, then this version of FGD will not work.* The FGD format is:

ANI*DNIS

Standard E&M Wink digit format is:

DNIS

FGD provides a method of sending CID information across a trunk that traditionally did not support CID. *Note that CID name is not available.* Caller ID to a FGD trunk can be debugged in the AOS voice device by the following command:

debug voice toneservices

A typical debug looks like this:

```
15:12:44 TONESERVICES.EVENTS fxs 0/2 - empty - Tone Detection: Request resource
15:12:44 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Tone Detection: DSP channel allocated for the resource
15:12:44 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Tone Detection: constructed
15:12:44 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Tone Detection: starting
15:12:44 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Tone Detection: TDM map
15:12:44 TONESERVICES.EVENTS fxs 0/2 - empty - DialTone Generation: Request resource
15:12:44 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - DialTone Generation: DSP channel allocated for the resource
15:12:44 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - DialTone Generation: constructed
15:12:44 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - DialTone Generation: starting
15:12:44 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - DialTone Generation: TDM map
15:12:46 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Tone Detection: received digit (5) event
15:12:46 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - DialTone Generation: stopping
15:12:46 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - DialTone Generation: TDM unmap
15:12:46 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - DialTone Generation: release
15:12:47 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Tone Detection: received digit (1) event
15:12:47 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Tone Detection: received digit (0) event
15:12:47 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Tone Detection: received digit (0) event
15:12:47 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Tone Detection: stopping
15:12:47 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Tone Detection: TDM unmap
15:12:47 TONESERVICES.EVENTS fxs 0/2 - dsp 0/2.1 - Tone Detection: release
15:12:48 TONESERVICES.EVENTS t1 0/4.0.2 - empty - DTMF Generation: Request resource
15:12:48 TONESERVICES.EVENTS t1 0/4.0.2 - dsp 0/2.1 - DTMF Generation: DSP channel allocated for the resource
15:12:48 TONESERVICES.EVENTS t1 0/4.0.2 - dsp 0/2.1 - DTMF Generation: constructed
15:12:49 TONESERVICES.EVENTS t1 0/4.0.2 - dsp 0/2.1 - DTMF Generation: sending DTMF digits (*9077*5100*)
15:12:49 TONESERVICES.EVENTS t1 0/4.0.2 - dsp 0/2.1 - DTMF Generation: TDM map
15:12:50 TONESERVICES.EVENTS t1 0/4.0.2 - dsp 0/2.1 - DTMF Generation: stopping
15:12:50 TONESERVICES.EVENTS t1 0/4.0.2 - dsp 0/2.1 - DTMF Generation: TDM unmap
15:12:50 TONESERVICES.EVENTS t1 0/4.0.2 - dsp 0/2.1 - DTMF Generation: release
```

The highlighted line shows the DTMF digits that were sent.